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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/806,662	03/23/2004	Tadao Kikumoto	230980-0248	1164

23392 7590 10/01/2009

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EXAMINER	
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ART UNIT	PAPER NUMBER
2626	

MAIL DATE	DELIVERY MODE
10/01/2009	PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 10/806,662	Applicant(s) KIKUMOTO, TADAO	
	Examiner ERIC YEN	Art Unit 2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.
- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 13 July 2009.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-50 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-50 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Response to Amendment

1. In response to the Office Action mailed 4/15/09, applicant has submitted an amendment filed 7/13/09.

New Claims 49-50 have been added.

Response to Arguments

1. Applicant's arguments filed 7/13/09 have been fully considered but they are not persuasive.

As per Claims 1 and 37-38, Applicant argues that the combination of Choi, Cano, and Gibson do not teach "the respective center frequencies of which have been fixed" and "setting modulation levels at the fixed center frequency of each of the frequency bands" (Amendment, page 11).

Gibson, however, teaches performing voice changing by splitting a signal into a high-band and a low-band. Frequency bands are defined at some point and cover a range of frequencies. By defining the range of frequencies a frequency band covers, the frequencies in the band are "fixed". Since this range has a middle value (e.g., a frequency band from 0 Hz to 10 Hz has a middle value of 5 Hz), by fixing/defining the range of frequencies a frequency band covers, each and every frequency within the band is fixed, including the middle value. This middle value is a "center frequency" in the frequency band. Therefore, by determining what range a frequency band covers,

Gibson teaches fixed center frequencies. These center frequencies are fixed by either the designer that specifies the frequency band or the system which specifies/fixes the frequency band whenever it elects to divide the signal for band-specific processing.

Also, since Choi describes modulation as the changing of one voice into another, by changing the spectral envelope of a frequency band (as in Gibson), the frequencies within the envelope are changed to a different amplitude (i.e., modulation level, since the new amplitude is a changed ["modulated" according to Choi's interpretation of modulation] amplitude, and the amplitude is a level for a particular frequency component), which changes <u>all</u> amplitudes and/or other frequency characteristics <u>included in the frequency band</u>. If the amplitude, for example, of the middle frequency (i.e., center frequency) of the frequency band is changed during voice changing (i.e., modulation) to a different amplitude defined by the system (e.g., the interpolated value between the source and target singer), the amplitude of this middle frequency is set/changed to a new level that accomplishes voice changing/modulation. Therefore, Gibson does teach/suggest where the respective center frequencies are fixed and where a new modulation level is set at the center frequency.

Applicant does not claim that <u>only</u> the center frequency's modulation level is set. Therefore, even if Gibson does teach changing values at other frequencies, the modification of spectral information in the entire frequency band reads on applicant's claim language as long as the center frequency's component <u>is changed too</u> as a result of the processing of the frequency band. Applicant does not claim that the frequency bands are defined by first specifying what the center frequencies are, or anything else

that prevents the defining of a frequency band, generally, from reading on applicant's claim language.

Applicant first argues that it would not have been obvious to combine Choi and Cano because "there is no apparent reason to combine systems directed towards very different applications". Applicant states that Choi is directed to modulating a voice over a mobile terminal and Cano is directed to a Karaoke application, and further argues that Cano discloses that functions that are unique to a karaoke application are practically essential, including knowing words beforehand and containing other information for phonetic alignment in order to change one voice into another (Amendment, page 12).

However, applicant's argument is flawed because there is nothing about a mobile terminal that would prevent it from doing what a karaoke application does as well. Mobile terminals are known to have less processing power and storage than servers, however it is also well-known to allocate different functions to servers if the client does not have enough power.

More importantly, both applications in Choi and Cano are directed to changing a source voice into a target voice. Choi changes a source voice into a target voice <u>using a reference spectrum parameter</u> <u>from memory</u> which means that Choi's voice conversion, while implemented in a mobile terminal, can handle the storage of target data in order to change a source voice into a target voice. Therefore, Choi and Cano are also describing similar voice processing functions since they convert voices using

pre-stored target data. Therefore, applicant's argument that Choi and Cano are "systems directed towards very different applications" is not true.

Just because Choi does not describe phonetic recognition alignment which are, as applicant argues, "practically essential", this does not mean that the application in Choi can not be modified to include the phonetic recognition/alignment in order to handle singing voices. The combination of Choi and Cano can be made by performing a simple substitution of one system's voice converter with another, or by running both processes in parallel and allocating speaking voices to Choi's system and singing voices to Cano's system. There is nothing that requires the prior art processes to somehow be mixed together and there is nothing in Choi that would make it impossible or impractical to implement any other essential processes needed to convert a singing voice. Applicant's argument that Cano states that it would be impossible/impractical to convert voices without knowing lyrics, etc., beforehand would only carry weight if it were somehow impossible/impractical <u>for Choi to implement these processes</u>. The fact that Choi simply does not <u>disclose</u> such processes does <u>not</u> mean that Choi's system is <u>incapable</u> of performing such processes. Since Choi discloses changing voices using pre-stored data already (paragraph 47), Choi's system is capable of handling a process that uses pre-stored data. Also, just because Choi's described primary embodiment is a mobile terminal does not mean that Choi's system cannot be used in a system with more power.

Therefore, Cano and Choi are <u>not</u>, in fact, "so different from each other" as applicant argues (Amendment, page 12)

Choi teaches changing voices generally, but does not specifically teach musical signals. Since Cano teaches changing singing voices and singing voices are musical, the combination of Choi and Cano teach changing musical signals.

Applicant then argues that Gibson does not teach dividing signals into frequency bands with fixed center frequencies and setting modulation levels based on formant characteristics and formant control information and modulating levels based on the modulation level set by the setting means, and also argues that there is no apparent reason to combine Choi with Cano and Gibson (Amendment, page 13).

Applicand argues that this is because Gibson teaches "changing frequencies of spectral envelopes" (Amendment, page 14).

However, the claim language does not require that the amplitude and every value associated with that center frequency is what is being fixed. A reasonable alternative interpretation is that the center frequency itself is at a fixed location. For example, in a frequency band spanning 0 Hz and 10 Hz, the center frequency is unavoidably located (i.e., fixed) at 5Hz because that is the middle value in the band. Therefore, the fact that frequencies are changed (i.e., the amplitudes at the frequencies within a frequency band are changed) does not change the fact that, in Gibson, the center frequency of a frequency band is fixed at a single location, which reads on applicant's claim language.

Applicant then describes Gibson's five specific methods including a third method involving "shifting a spectral envelope in frequency", a first method involving "modifying the transfer function of the filter" which changes "coefficients of the filter in order to

change the frequencies of the spectral envelope", a second method "modifying the location of these singularities to generate a new digital filter having the desired spectral characteristics" and the fourth and fifth methods which applicant understands to also include changing spectral envelopes (Amendment, pages 14-15).

However, this does not mean that the combination of Choi, Cano and Gibson does not teach the recited claim language.

First, applicant's argument is flawed because changing the spectral envelope of a signal does not mean that the center frequency of the analysis frequency band is changed. Even if the envelope within a frequency band is shifted such that the frequency values previously located at frequencies within the band are shifted to frequencies out of the band, that does not mean that the frequency band itself changes location. It only means that the amplitude values in, for example, the low frequency band, are now shifted into the high frequency band, Or a lowER frequency band. The low frequency band specified for analysis, itself, still covers the range of frequencies specified for analysis. The low analysis band does not change in location just because the envelope that was original there has relocated to a frequency outside of that band.

Also, just because the envelope was shifted does not mean that the envelope is shifted outside of the frequency band. A formant can have an envelope covering the range 1 Hz to 2 Hz in a frequency band spanning 0 Hz to 10Hz. This envelope can be shifted to 8 Hz to 9 Hz, for example, changing the frequency of the sound, but it still falls within the 0 Hz to 10 Hz band. Therefore, it is not necessarily the case that shifting the spectral envelope necessarily changes the frequency band.

Even if spectral characteristics of an audio signal are changed, the analysis frequency band itself does not change its location. Therefore, the frequencies within the frequency band are still fixed, and this includes the middle frequency in the frequency band. Therefore, as long as the frequency band does not move, the teaching of a frequency band reads on applicant's "fixed center frequency".

Additionally, there is no requirement that the specific methods in Gibson must be incorporated into the combination if any part of Gibson is applied to a combination. Gibson was applied to teach only that low and high frequency bands in a voice changing application are processed separately. Therefore, even if there was some sort of inherent feature of Gibson's processing methods that requires changing the location of the low and high frequency bands, this does affect whether the combination reads on the claim language because the combination only includes the separate frequency band processing concept.

Gibson explicitly teaches that processing high and low frequencies separately reduces computation demands (col. 9, lines 62-65).

Therefore, it is not true, as applicant argues, that there is "no apparent reason" to combine Gibson with Choi and Cano.

Applicant then argues that Gibson, like Cano pertains to a karaoke device that requires recording a target voice and therefore cannot be combined. As discussed above, however, Choi, Cano, and Gibson all relate to voice changing based on pre-

stored data, and there is no requirement that Choi's device must be implemented on a mobile terminal or other device that cannot handle Cano or Gibson's voice changing processes. Therefore, their processes are analogous and not "so different" as applicant argues.

Therefore, the examiner maintains the previous prior art rejections restated below.

Regarding Claims 49-50, applicant argues that because Gibson teaches methods that involve changing filter coefficients, Gibson does not teach or suggests "free of changing the filter coefficients" (Amendment, page 17).

While this may be true, as discussed above, there is no requirement that just because Gibson teaches the concept of processing two frequency bands separately, that the two frequency bands must be processed using the methods of Gibson.

Cano and Choi each teach methods that modify prosody values (frequency, pitch, etc.) and not filter coefficients. Therefore, since they do not apply filter coefficients, the bandpass/highpass/lowpass filters used to divide the signals need not be changed. Low/high/band-pass filtering does not depend on the frequency analysis itself. Adaptive filters do exist in the art, but they are not mandatory. Nor are the processing methods of Gibson necessarily incorporated into the combination. Cano and Choi's voice changing methods can be simply substituted for the Gibson methods, thus excluding the Gibson methods from changing the division filters. Gibson also

teaches an example where one frequency band is greater than 10kHz and the other is less than or equal to 10 kHz. Combining this filtering with either Cano or Choi's method performs a voice changing process that does not affect the band division filters used to separate a high frequency component from a low frequency component for separate processing.

Furthermore, the methods described in columns 7-8 of Gibson do not modify the band division filters. The adaptive filters apply to transforming of the signal after the high/low-pass filtering. The part of Gibson that read on the "filter" of claim 4 was the high-pass and low-pass filters in, for example, Figure 8. These band division filters are not affected by the methods described in columns 7-8 because they apply to the "modify spectral envelope" sections of Figures 7-8. The spectral envelope modification relates to voice changing while the low and high pass filters are simply pre-filtering operations independent of the actual changing of the input voice.

Therefore, Gibson's description also reads on applicant's new claim 49 because filters are necessarily/obviously defined by their filter coefficients including the low and high pass filters and these filters are not changed based on the voice changing methods in Gibson.

Similarly as per new Claim 50, since the voice changing methods of Gibson don't affect the high/low-pass filters in Gibson and so the high/low-pass filters have coefficients that remain constant.

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1-50 are rejected under 35 U.S.C. 103(a) as being unpatentable over Choi (US 2003/0014246), in view of Cano et al. ("Voice Morphing System for Impersonating in Karaoke Applications"), hereafter Cano, and Gibson et al. (US 6,336,092).

As per Claim 1, Choi teaches a vocoder system comprising: formant detection means for analyzing a first tone signal to detect formant characteristics of the first tone signal ("voice signal of the subscriber… detect the spectrum parameter", paragraph 46; "spectrum parameter… are detected", paragraph 47; where the spectrum of a signal comprises, among other things, the formants of a voice)

tone signal input means for inputting a second tone signal that corresponds to specified pitch information ("selects the kind of the effect... converts the spectrum parameter… with reference to the loaded spectrum parameter…conversion of the spectrum parameter... height of voice", paragraph 47)

setting means for setting modulation levels based on the formant characteristics and formant control information with which the formant characteristics detected by the formant detection means are changed ("selects the kind of the effect... converts the

spectrum parameter… with reference to the loaded spectrum parameter…conversion of the spectrum parameter... height of voice”, paragraph 47; “modulating”, paragraph 38)

modulation means for modulating a level of a signal based on the modulation level set in the setting means (“modulating”, paragraph 38).

Choi fails to teach the tone signals are musical tone signals.

Cano teaches the tone signals are musical tone signals ("target singer", Introduction).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi to include the teaching of Cano of the tone signals are musical tone signals, in order to extend voice changing to singing applications, as described by Cano (Introduction).

Choi, in view of Cano, fail to teach division means for dividing the second musical tone signal into a plurality of frequency bands, the respective center frequencies of which have been fixed, where the modulation levels are set at the fixed center frequency of each of the frequency bands, and where modulating the level of a signal modulates levels of each of the frequency bands.

Gibson suggests division means for dividing the second musical tone signal into a plurality of frequency bands, the respective center frequencies of which have been fixed, where the modulation levels are set at the fixed center frequency of each of the frequency bands, and where modulating the level of a signal modulates levels of each of the frequency bands (“signal is split into two equal-width frequency bands… gain compensation… transformed ”, col. 9, lines 44-65; “summing a gain-compensated high-

frequency signal and the transformed low-frequency component", col. 9, line 65 – col. 10, line 2; "source and target voice signals", col. 7, lines 17-28; where, to determine the target voice characteristics and the necessary transformations, an analysis of the target voice signals in the corresponding frequency bands is obvious/necessary).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of division means for dividing the second musical tone signal into a plurality of frequency bands, the respective center frequencies of which have been fixed, where the modulation levels are set at the fixed center frequency of each of the frequency bands, and where modulating the level of a signal modulates levels of each of the frequency bands, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claims 37-38, their limitations are similar to those in Claim 1, and so are rejected under similar rationale.

As per Claim 2, Choi fails to teach wherein the format detection means comprises a filter.

Gibson suggests wherein the format detection means comprises a filter ("signal is split into two equal-width frequency bands… gain compensation… transformed ", col. 9, lines 44-65; "summing a gain-compensated high-frequency signal and the transformed low-frequency component", col. 9, line 65 – col. 10, line 2; "source and

target voice signals", col. 7, lines 17-28; where division into frequency bands involves filtering an input spectrum).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of wherein the format detection means comprises a filter, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claim 3, Choi teaches wherein the formant detection comprises a Fourier transform ("spectrum parameter… are detected", paragraph 47; where a spectrum is a frequency domain representation obtained by applying a transform, which is commonly a Fourier transform).

As per Claims 4-6, Choi, in view of Cano, fail to teach wherein the division means comprises a filter.

Gibson suggests wherein the division means comprises a filter ("signal is split into two equal-width frequency bands… gain compensation… transformed ", col. 9, lines 44-65; "summing a gain-compensated high-frequency signal and the transformed low-frequency component", col. 9, line 65 – col. 10, line 2; "source and target voice signals", col. 7, lines 17-28; where division into frequency bands involves filtering an input spectrum).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of

wherein the division means comprises a filter, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claim 49, Choi, in view of Cano, fail to teach wherein the filter comprises a digital filter having frequency characteristics defined by a plurality of filter coefficients and wherein the setting means sets the modulation levels, free of changing the filter coefficients.

Gibson teaches the filter comprises a digital filter having frequency characteristics defined by a plurality of filter coefficients and wherein the setting means sets the modulation levels, free of changing the filter coefficients (Figures 7-8, low pass and high pass filters; "method ... modify the original spectral envelope", col. 7, line 18 – col. 8, line 28; see Response to arguments, where all filters including the low and high pass filters at least obviously are defined by their filter coefficients [based on mathematical theory], and also the filters being modified by the methods are applied in the "modify... spectral envelope" blocks which are subsequent processes independent of the high and low pass filters themselves, and so the high and low pass filter coefficients are not changed).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of the filter comprises a digital filter having frequency characteristics defined by a plurality of filter coefficients and wherein the setting means sets the modulation levels, free of

changing the filter coefficients, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claim 50, Choi, in view of Cano, fail to teach wherein the filter comprise a digital filter having frequency characteristics defined by a plurality of filter coefficients, and wherein the setting means sets the modulation levels while the filter coefficients remain constant.

Gibson teaches wherein the filter comprise a digital filter having frequency characteristics defined by a plurality of filter coefficients, and wherein the setting means sets the modulation levels while the filter coefficients remain constant (Figures 7-8, low pass and high pass filters; “method … modify the original spectral envelope”, col. 7, line 18 – col. 8, line 28; see Response to arguments, where all filters including the low and high pass filters at least obviously are defined by their filter coefficients [based on mathematical theory], and also the filters being modified by the methods are applied in the "modify... spectral envelope" blocks which are subsequent processes independent of the high and low pass filters themselves, and so the high and low pass filter coefficients are not changed).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of wherein the filter comprise a digital filter having frequency characteristics defined by a plurality of filter coefficients, and wherein the setting means sets the modulation levels

while the filter coefficients remain constant, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claims 7-9, Choi, in view of Cano, fail to teach wherein the division means comprises a Fourier transform.

Gibson suggests wherein the division means comprises a Fourier transform ("signal is split into two equal-width frequency bands… gain compensation… transformed", col. 9, lines 44-65; "summing a gain-compensated high-frequency signal and the transformed low-frequency component", col. 9, line 65 – col. 10, line 2; "source and target voice signals", col. 7, lines 17-28; "spectral", col. 9, lines 32-43; where division into frequency bands involves filtering an input spectrum).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of wherein the division means comprises a Fourier transform, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claims 10-18, Choi fails to teach wherein the setting means sets the modulation levels by interpolation processing based on the formant characteristics and the formant control information.

Cano teaches wherein the setting means sets the modulation levels by interpolation processing based on the formant characteristics and the formant control information ("target singer", Introduction; “interpolated”, Section 2).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi to include the teaching of Cano of wherein the setting means sets the modulation levels by interpolation processing based on the formant characteristics and the formant control information, in order to extend voice changing to singing applications, as described by Cano (Introduction).

As per Claim 19-23, Choi teaches wherein the setting means sets modulation levels based on pitch information, the formant characteristics, and the formant control information (“pitch… pitch period”, paragraph 35; “converting… the pitch period”, paragraph 17).

As per Claims 24-27, Choi teaches wherein the setting means sets modulation levels based on musical interval, the formant characteristics, and the formant control information (“pitch… pitch period”, paragraph 35; “converting… the pitch period”, paragraph 17; where a “period” is an interval, and the pitch applies to musical characteristics).

As per Claim 28-36, Choi teaches wherein the setting means stores a formant change table that changes the formant non-uniformly and sets the modulation levels

based on the change table ("selected effect", paragraph 19; "cave", paragraph 47; where the information for each of the effects must be arranged in memory to be found by the application, and this organized memory is a form of table).

Choi, in view of Cano, fail to teach where the modulation levels correspond to each of the frequency bands.

Gibson suggests where the modulation levels correspond to each of the frequency bands (“signal is split into two equal-width frequency bands… gain compensation… transformed”, col. 9, lines 44-65; “summing a gain-compensated high-frequency signal and the transformed low-frequency component”, col. 9, line 65 – col. 10, line 2; “source and target voice signals”, col. 7, lines 17-28; “spectral”, col. 9, lines 32-43; where division into frequency bands involves filtering an input spectrum).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of where the modulation levels correspond to each of the frequency bands, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claim 39, its limitations are similar to those in Claim 2, and so is rejected under similar rationale.

As per Claim 40, its limitations are similar to those in Claim 3, and so is rejected under similar rationale.

As per Claim 41, Choi teaches wherein the first musical tone signal is produced by a male voice or a female voice ("voice", paragraph 4; where voices by a human are either male or female).

As per Claim 42, Choi, in view of Cano, fail to teach wherein the level of the signal of each of the frequency bands modulated by the modulation means is an amplitude of the signal.

Gibson suggests division wherein the level of the signal of each of the frequency bands modulated by the modulation means is an amplitude of the signal ("signal is split into two equal-width frequency bands… gain compensation… transformed ", col. 9, lines 44-65; "summing a gain-compensated high-frequency signal and the transformed low-frequency component", col. 9, line 65 – col. 10, line 2; "source and target voice signals", col. 7, lines 17-28; gains affect amplitudes of a spectrum).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of wherein the level of the signal of each of the frequency bands modulated by the modulation means is an amplitude of the signal, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claim 43, Choi, in view of Cano, fail to teach wherein, in the modulation means, the center frequencies of the frequency bands are maintained as fixed in the division means.

Gibson suggests wherein, in the modulation means, the center frequencies of the frequency bands are maintained as fixed in the division means ("signal is split into two equal-width frequency bands... gain compensation... transformed ", col. 9, lines 44-65; "summing a gain-compensated high-frequency signal and the transformed low-frequency component", col. 9, line 65 – col. 10, line 2; "source and target voice signals", col. 7, lines 17-28; the filters do not change the frequency range that they occupy, and so their center frequencies do not change either).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of wherein, in the modulation means, the center frequencies of the frequency bands are maintained as fixed in the division means, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claims 44, Choi fails to teach wherein the setting means sets the modulation levels by using a polynomial interpolation.

Cano teaches wherein the setting means sets the modulation levels by using a polynomial interpolation ("target singer", Introduction; "interpolated", Section 2; where the use of polynomial interpolations are an obvious to one of ordinary skill in the art as a type of interpolation that can be used to convert voices).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi to include the teaching of Cano of wherein the setting means sets the modulation levels by using a polynomial interpolation, in order to extend voice changing to singing applications, as described by Cano (Introduction).

As per Claim 45, Choi, in view of Cano, fail to teach wherein the center frequencies of the modulated signals of the frequency bands are equal to the respective center frequencies of the frequency bands, as fixed by the division means.

Gibson suggests wherein the center frequencies of the modulated signals of the frequency bands are equal to the respective center frequencies of the frequency bands, as fixed by the division means ("signal is split into two equal-width frequency bands… gain compensation… transformed ", col. 9, lines 44-65; "summing a gain-compensated high-frequency signal and the transformed low-frequency component", col. 9, line 65 – col. 10, line 2; "source and target voice signals", col. 7, lines 17-28; the filters do not change the frequency range that they occupy, and so their center frequencies do not change either).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of wherein the center frequencies of the modulated signals of the frequency bands are equal to the respective center frequencies of the frequency bands, as fixed by the division means, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claim 46, Choi teaches wherein the first musical tone signal is a speech signal ("voice", paragraph 4).

As per Claim 47, Choi fails to teach wherein the setting means sets the modulation level by interpolation processing based on the formant characteristics at a plurality of frequencies.

Cano teaches wherein the setting means sets the modulation level by interpolation processing based on the formant characteristics at a plurality of frequencies ("target singer", Introduction; "interpolated", Section 2; where the use of polynomial interpolations are an obvious to one of ordinary skill in the art as a type of interpolation that can be used to convert voices).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi to include the teaching of Cano of wherein the setting means sets the modulation level by interpolation processing based on the formant characteristics at a plurality of frequencies, in order to extend voice changing to singing applications, as described by Cano (Introduction).

Choi, in view of Cano, fail to teach the modulation level is set at the fixed center frequency of at least one of the frequency bands.

Gibson suggests the modulation level is set at the fixed center frequency of at least one of the frequency bands ("signal is split into two equal-width frequency bands… gain compensation… transformed ", col. 9, lines 44-65; "summing a gain-compensated

high-frequency signal and the transformed low-frequency component", col. 9, line 65 – col. 10, line 2; "source and target voice signals", col. 7, lines 17-28; the filters do not change the frequency range that they occupy, and so their center frequencies do not change either).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of the modulation level is set at the fixed center frequency of at least one of the frequency bands, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

As per Claims 48, Choi fails to teach wherein the setting means sets the modulation levels by using a polynomial interpolation of the formant characteristics at a plurality of frequencies.

Cano teaches wherein the setting means sets the modulation levels by using a polynomial interpolation of the formant characteristics at a plurality of frequencies ("target singer", Introduction; "interpolated", Section 2; where the use of polynomial interpolations are an obvious to one of ordinary skill in the art as a type of interpolation that can be used to convert voices).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi to include the teaching of Cano of wherein the setting means sets the modulation levels by using a polynomial interpolation, in order to extend voice changing to singing applications, as described by Cano (Introduction).

Choi, in view of Cano, fail to teach wherein the modulation levels are set at the fixed center frequency of at least one of the frequency bands.

Gibson suggests wherein the modulation levels are set at the fixed center frequency of at least one of the frequency bands ("signal is split into two equal-width frequency bands... gain compensation... transformed ", col. 9, lines 44-65; "summing a gain-compensated high-frequency signal and the transformed low-frequency component", col. 9, line 65 – col. 10, line 2; "source and target voice signals", col. 7, lines 17-28; the filters do not change the frequency range that they occupy, and so their center frequencies do not change either).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Choi, in view of Cano, to include the teaching of Gibson of wherein the modulation levels are set at the fixed center frequency of at least one of the frequency bands, in order to transform voices with reduced computational demands, as described by Gibson (col. 9, lines 62-65).

Conclusion

3. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL.** See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within

TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to ERIC YEN whose telephone number is (571)272-4249. The examiner can normally be reached on M-F 7:30-4:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on 571-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

EY 9/28/09

/Richemond Dorvil/
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